Reference for Nortel DMS-100 CO Switch

Step 8: Starting CallPath Server for OS/2 and the DMS-100 CO switch connection or connections

Following is an overview of the tasks required for starting the CallPath Server for OS/2 and the DMS-100 CO switch connection or connections:

- Start the CallPath Server Management Facility
- Start the CallPath Server for OS/2 subsystem
- Start the DMS-100 CO switch connection or connections

These tasks are described in more detail in the following sections.

If you encounter errors or problems, refer to *CallPath Server Planning, Installation, and Problem Determination Guide.*

Starting the CallPath Server Management Facility

To start the CallPath Server Management Facility, do *one* of the following:

- Type `csasrvr` at the OS/2 command prompt.
- Double-click the **CallPath** icon on the OS/2 desktop and then double-click the **CallPath Server Management Facility** icon.

After CallPath Server for OS/2 is started, the CallPath Server Management Facility window and the Product Information window are displayed.

If password protection was enabled, the Administrative Password window is also displayed and the password must be entered to gain access to the CallPath Server Management Facility configuration functions. To enter the password and gain access to the CallPath Server Management Facility, type the password, then click **OK**.

Note:

The default password, until you change it, is *password.*

Starting the CallPath Server for OS/2 subsystem

After starting the CallPath Server Management Facility, start the CallPath Server for OS/2 subsystem in *one* of the following ways:

- From the CallPath Server Management Facility window:
  1. Select the **Administration** menu from the menu bar.
  2. Select the **Server** menu option.
  3. Click **Start**.
  4. Click on *one* of the following in response to the confirmation message:

   - **Yes** to start the CallPath Server for OS/2 subsystem
   - **No** to return to the CallPath Server Management Facility window without starting the CallPath Server for OS/2 subsystem
5. Click **Yes** on the confirmation message window.

- From the **CallPath Server** icon:
  1. Click the **CallPath Server** icon.
  2. Click **Start**.
  3. Click on *one* of the following in response to the confirmation message:
     - **Yes** to start the CallPath Server for OS/2 subsystem
     - **No** to return to the CallPath Server Management Facility window without starting the CallPath Server for OS/2 subsystem

Starting one or more DMS-100 CO switch connections

After starting the CallPath Server for OS/2 subsystem, start your DMS-100 CO switch connection in *one* of the following ways:

- From the CallPath Server Management Facility window menu bar:
  1. Select the **Administration** menu.
  2. Select the **Switch Connection** menu option.
  3. Click **Start**.
  4. Select the switch.
  5. Click on *one* of the following in response to the confirmation message:
     - **Yes** to start the switch
     - **No** to return to the CallPath Server Management Facility window without starting the switch
- From the CallPath Server Management Facility window switch icons:
  1. Click the Switch icon.
  2. Click **Start**.
  3. Click on *one* of the following in response to the confirmation message:
     - **Yes** to start the switch
     - **No** to return to the CallPath Server Management Facility window without starting the switch

Repeat one of the above procedures for each switch connection you want to start.

Multiple logons using the same service ID

Multiple logons to a single business group using the same service ID cause unpredictable results and are not supported by this program. If multiple logons occur, the switch response to a request is sent to the next available server link instead of being sent to the server making the request. This causes the call state model to be out of synchronization and the server stops due to a critical error.

To avoid such an error, log on to a business group using the same service ID for only one switch link at a time. This means that any switch session logon to a given business group using a given service ID must be stopped before a different logon to the same business group using the same service ID can be attempted.

For example, switch session **SWITCH01** and switch session **SWITCH02** each have the same password and the same service ID and both are enabled to log on to the same business group. If switch session

SWITCH01 is active, but you want to start an application that uses switch session **SWITCH02**, switch session **SWITCH01** must be stopped first. This is true even if switch session **SWITCH01** has no traffic. Switch links **SWITCH01** and **SWITCH02** cannot be simultaneously active.

Note:

This limitation is true for any combination of SwitchServer/2, CallPath Server for OS/2, CallPath Server for AIX, and CallPath Server for Windows NT/Windows 2000 link sessions that can log on to the same business group. It is recommended that switch sessions avoid overlapping.

Multiple logons using different service IDs

The switch can be configured to enable independent logons to a business group using different service IDs for each logon. The switch considers these to be independent sessions. The *first* session to execute an association (monitoring) of a primary ACD-DN or Centrex DN has exclusive event messaging for that resource. All events concerning that resource remain within the link established during that session. If a session ends for any reason (such as the failure of an X.25 link) any remaining session can execute an association with the orphaned resource.

The switch enables up to eight sessions for each business group. Each session requires a unique service ID. This process partitions a business group into unique and independent sub-business groups. The Server supports this independent, simultaneous, switch session environment.

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SIP CENTER

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Nortel has fully embraced the Session Initiation Protocol (SIP) for delivering next-generation multimedia communication services.

Nortel is an industry leader and innovator removing barriers to efficiency, productivity and growth. Leveraging our broad technology expertise, sophisticated network know-how, and large installed base of service provider and enterprise customers, we're transforming networks to enrich global communications. Nortel Networks is creating innovative, packet based networks that are more robust, intelligent and secure, and provide growth opportunities for our customers by enabling new services and applications.

Nortel enables new revenue generation opportunities, operating cost reductions, enhanced customer services and improved productivity by leveraging our innovative technology solutions and services. We can help make your business a more profitable place.

Today, we're delivering networking and communication services and infrastructure to service providers and enterprises in more than 150 countries. Customers in the United States, Europe, Asia-Pacific, Caribbean and Latin America, the Middle East, Africa, and Canada benefit from our commitment to technology leadership and our culture of innovation.

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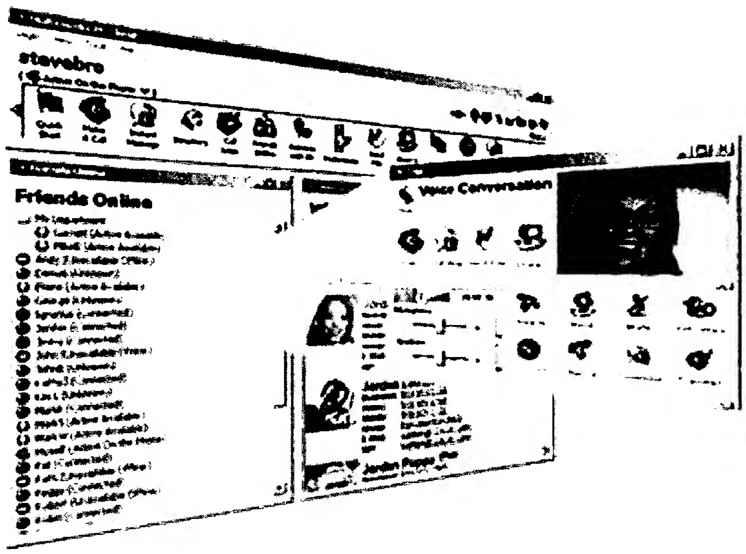

Multimedia Communications Portfolio
Nortel Multimedia Communications portfolio includes carrier-hosted and enterprise multimedia services solutions that unleash communications simplicity and productivity for enterprise and residential subscribers. This portfolio leverages the emerging industry standard Session Initiation Protocol (SIP) to deliver multimedia services including voice, call management, desktop video calling, collaboration tools, and personalization services. More >>>

Multimedia Communication Server 5100
The Nortel Multimedia Communication Server (MCS) 5100 is a multimedia communications and collaboration applications delivery solution for Enterprises that seamlessly delivers converged multimedia services to employees regardless of their work location. The MCS 5100 drives VoIP networks, delivering multimedia and collaborative applications to the enterprise on an "open" platform that supports industry-standard protocols, including SIP and H.323. More >>>

Multimedia Communication Server 5200
The Nortel Multimedia Communication Server 5200—which seamlessly integrates voice with video, collaboration, and presence services—enables service providers to deliver hosted multimedia communications services to create new revenue-generating opportunities. These Session Initiation Protocol-based (SIP), next generation, multimedia services enhance the ability for both enterprise and consumer end-users to communicate and collaborate more effectively and efficiently. More >>>

Click here for a flash overview of the MCS 5200 >>>

Communication Server 1000
The Nortel Communication Server 1000 is a full-featured IP telephony solution for the medium to large enterprise environment (40-10,000+ users) that provides robust, survivable, scalable IP-based telephony services capable of being distributed across IP Wide Area Networks (WANs) delivering a full range of proven applications. More >>>

Communication Server 2100
The Nortel Communication Server 2100 is a full-featured IP telephony solution for the large to campus enterprise environment (2,000-200,000 users) that provides robust, survivable, scalable IP-based telephony services capable of being distributed across IP Wide Area Networks (WANs) delivering a full range of proven applications.

Communication Server 2000 - Compact
The Nortel Communication Server 2000 - Compact is a scalable, reduced-footprint, Superclass softswitch built on the commercially-available Compact PCI platform. Delivering a comprehensive set of Class 4 & 5 features, this carrier-grade softswitch is applicable for new network builds and packet evolution. Based on open hardware and software and supporting industry-standard protocols, it is ideal for Service Providers and Cable Multiple Systems Operators. More >>>

Alteon Application Switches
Nortel Alteon Application Switches enable application optimization, delivery, and high availability through the use of sophisticated application/device load balancing, intelligent traffic management, application redirection, application-layer security, security acceleration, and bandwidth management. For example, in a VoIP (SIP) call server optimization scenario, Alteon Application Switches can dynamically distribute load among multiple SIP Proxy servers using SIP Call ID information. This solution enables resilient VoIP services by ensuring call processing resources are 'always up', capacity additions can be made without downtime, and call traffic is distributed across all call servers to optimize performance and utilization. More >>>

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3com

3Com offers a unique blend of practical and innovative technology that provides its customers with high-value, practical and economic solutions.

More than twenty years ago 3Com pioneered local area networks with the industry's first Ethernet network interface cards. Over the past two decades, the company has earned more than 917 U.S. patents--with over 873 additional applications pending--for a broad range of networking solutions that have been integrated into a range of prizewinning local- and wide-area network products.

In 1998 3Com initiated another landmark innovation: carrier-class IP telephony--adopted by several major telecommunications carriers with a record now of more than 20 billion minutes of calls. In that same year 3Com marketed the first

networked telephony solutions to businesses, the 3Com NBX® systems, which have been deployed in over 15,000 locations and have garnered more than 40 industry awards. Building on this experience, today 3Com offers carrier-class IP telephony to enterprises of all sizes with the VCX V7000 IP Telephony Solution. This comprehensive, exceptionally flexible, reliable, and manageable voice communications system can increase business productivity, improve customer relationships, and reduce costs at a time when businesses more than ever must leverage their infrastructure investments to remain competitive.

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ACE*COMM

ACE*COMM is the global leader in advanced Convergent Mediation ™ products for wired and wireless voice, data, and Internet communications providers. Our proven technology enables the capture, security, validation, correlation, augmentation, and warehousing of data from all network elements and distributes it in appropriate formats to all types of operations and business support systems (OSS/BSS). Our solutions may be tailored to each customer's needs, providing the analytical tools to extract knowledge from their networks-knowledge they use to reduce costs, accelerate time-to-market for new products and services, generate new sources of revenue, and push forward with next-generation initiatives. ACE*COMM technology has been successfully deployed in over 3500 installations in more than 65 countries worldwide.

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Brooktrout Technology®

Brooktrout is a leading supplier of enabling software and hardware, including platforms and media servers, which provide the media processing and call control functions for next generation IP-based applications and enhanced services. Brooktrout's products are used in wireline, wireless and broadband networks for next generation IP voice and data applications such as voice and video conferencing, messaging and prepaid services.

AG Projects

Focus

AG Projects provides managed services and solutions for Next Generation Networks since 2002.

Having a broad experience in several technologies and the networking in

between, AG Projects is a unique vendor able to provide most of the key infrastructure elements required to run Voice over IP. AG Projects provides cost-effective integrated solutions which you would normally purchase from multiple vendors.

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Your Missing Link™

Headquartered in New York City, Antepo is the only provider of Enterprise Instant Messaging (EIM) systems to offer multi-standard interoperability and enterprise reliability.

Antepo's Open Presence Network™ (OPN) solutions enable real-time communication and collaboration between employees, customers, suppliers, and partners, while meeting critical requirements for control, security, integration, and compliance. Key to Antepo's success is the company's clear advantage as the only EIM provider to deliver carrier-class reliability and scalability, and to support Presence aggregation and federation across both industry-standard protocols (XMPP and SIP/SIMPLE).

Antepo's clients include financial institutions, telecommunications companies, and globally diversified companies in the US and internationally.

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Convedia is a leading supplier of media servers, which allow telecommunication service providers to increase enhanced service revenues while reducing costs. Convedia's multi-protocol media servers the functions of traditional announcement servers, interactive voice response (IVR/VRU) units, conference bridges, messaging platforms and speech platforms into a multi-service, open standards compliant solution for wireline, wireless, and cable network operators. Convedia's award winning family of products includes the CMS-6000 Media Server, a carrier class media server capable of scaling to 18,000 ports per shelf and the CMS-1000 Media Server, a low cost, entry level media server designed for smaller service providers, large enterprise and network edge deployment.

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Independent developer of core systems technology for over 20 years, Data Connection Limited (DCL) supplies the leading players in the computer industry with their network protocol, conferencing and messaging software. Their success is based on technological expertise and the ability to delivery high quality, high-function software on time.
DCL provides a robust, scalable SIP implementation designed for carrier-class devices and which is built on the architecture and experience behind their MGCP, Megaco/H.248, SCTP, MPLS, IP Routing and ATM implementations. All their products are backed by their unrivalled support service.

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ERICSSON

Ericsson is a world-leading supplier in the fast growing and dynamic telecommunications and data communications industry, offering advanced communications solutions for mobile and fixed networks, as well as consumer products. Ericsson is a total solutions supplier for all customer segments: network operators and service providers, enterprises and consumers.

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GL Communications

GL Communications is a leading provider of test and measurement tools for TDM, VoIP, and Wireless networks. Our Packet Series™ products are PacketGen™, PacketScan™, RTP Toolbox™, and H.323 Call Emulator / Analyzer.

PacketGen™ is a SIP bulk call generator that can emulate thousands of SIP calls with real-time traffic over established calls, such as Voice, Digits, Tones, Noise, Live Speech (all codecs), and Fax. PacketGen™ is fully SIP compliant with a distributed architecture that permits modular scalability and remotability. PacketGen™ can be integrated with GL's Voice Quality Testing (VQT) product providing true voice quality of SIP sessions. VQT utilizes ITU-standard voice quality assessment algorithms (PESQ, PAMS, and PSQM).

PacketScan™ is a realtime protocol analyzer that can analyze thousands of SIP sessions in detail. Statistics include: packet delay, gap, jitter, and lost packets. PacketScan™ also permits realtime listening, recording, oscilloscope and spectral analysis of RTP signals in a variety of codecs including G.711, G.729AB, G.726, and GSM. PacketScan™ is a complete VoIP protocol analyzer with full decoce and analysis of H323, Sigtran, MGCP, Megaco, UDP, RTP, RTCP, and others. A related product RTPToolbox™ provides detailed emulation and analysis of RTP streams.

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Hotsip is a leading SIP Application Server provider with applications for large scale, carrier grade SIP enabled broadband and 3G/IMS networks. Hotsip offers convergent off-the-shelf applications as well as an open Service Creation Environment (SCE) for building new customized SIP and web applications.

Major client wins to date include TeliaSonera, Bell Net, WX3 and Tussa. Hotsip has also developed partnerships with leading industry vendors including Cisco, Ericsson, HP and Nokia.

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HP delivers more of what network, service and content providers demand - an adaptive enterprise that efficiently deploys personalized, content-rich services to drive profitable revenue. HP is a technology solutions provider to consumers, businesses and institutions globally. The company's offerings span IT infrastructure, personal computing and access devices, global services and imaging and printing for consumers, enterprises and small and medium businesses. For the last four quarters, HP revenue totaled $71.3 billion.

More information about the HP network and service providers industry vertical is available at www.hp.com/go/nsp

More information about HP is available at www.hp.com

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Interactive Intelligence Inc. (Nasdaq: ININ) is a global developer of software for SIP-based IP telephony, contact center automation and unified communications. The company was founded in 1994 by Dr. Donald E. Brown and employs approximately 350 people serving more than 1,000 customers worldwide.

Interactive Intelligence's Interaction Center Platform® is the foundation for its award-winning suite of Windows 2000-based IP telephony software products, which, in addition to providing built-in SIP support, extend IP PBX functionality to include multimedia routing and queuing, interactive voice response, voice mail, unified messaging, fax-on-demand, Internet text chat, Web callback and more.

This software is unique because, unlike proprietary IP telephony solutions, it enables organizations to deploy the IP network of their choice, while getting a wide range of SIP-enabled features all running on a single, open server for reduced costs, increased productivity and maximum investment protection.

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Lucent Technologies
Bell Labs Innovations

Lucent Technologies designs and delivers network products and software for the world's leading communications service providers. Innovative Bell Labs research and development continue to contribute to Lucent's strengths in mobility, optical, data and voice networking technologies as well as software and services to leverage and evolve existing investments to next-generation networks. Using powerful Bell Labs tools and Lucent Worldwide Services, we also offer professional services for revenue recovery, network optimization, inventory management and network security, and managed services. Our solutions are designed to help customers quickly deploy and better manage their networks and create new, revenue-generating services that help businesses and consumers. Lucent Technologies is headquartered in Murray Hill NJ, USA. For more information about Lucent Technologies, please visit:

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MITEL

Bringing SIP-based Services to the Desktop In the race to deliver new value-added broadband voice and multimedia SIP-based services to business, the finish line is at the desktop. And Mitel Networks is the only vendor that takes you all the way, with a complete portfolio of SIP phones, devices and peripherals that enable users to access and benefit from new and emerging SIP services and applications.

Mitel Networks is a leading-edge provider of next-generation IP communications solutions and a market-leader for voice video and data convergence. The company creates advanced communication solutions and applications in the areas of speech recognition, wireless mobility unified messaging and customer interaction solutions. Mitel integrates voice and data infrastructures with its patented dual-bus architecture in its Integrated Communications Platforms. Mitel Networks currently serves the education, hospitality, and government markets. The company is headquartered in Ottawa Canada. Mitel Networks is committed to the development of SIP- supported IP devices and platforms and recognizes the huge opportunity that the adoption of SIP presents to the enterprise IP telephony industry.

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NAVTEL

Navtel Communications provides high capacity testing solutions that enable telecom equipment manufacturers and network providers to verify under full load the characteristics and the design of triple play equipment and converged networks using a single, high performance, scalable platform - the InterWatch.

Navtel's InterWatch can substantially reduce both costs and timelines in the new product development process. It plays a major role in the rapid development and deployment of the next generation of converged networks around the globe by organizations who are secure in the knowledge that Navtel's range of load testing solutions are complimented by their unrivalled ability to also test for conformance, functionality and interoperability on the single most powerful integrated test platform available today.

Navtel has been a very early adopter of the SIP Technology. Its involvement in standard bodies and long-term expertise in the test equipment market has enabled it to develop a complete product portfolio for both manufacturers and carriers laboratories.

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NETRAKE

Netrake is the premier provider of session controllers delivering real-time control of voice and multimedia over IP networks. Utilizing its advanced network processing architecture, Netrake's nCite system provides unprecedented VoIP session control such as; multi-layer network translation, firewall traversal, quality of service mediation and session detail records. Netrake's combination of features, performance, and reliability give global service providers a scalable, cost-effective answer to enterprise and carrier VoIP interconnection. Netrake is a privately held company, headquartered in Plano, Texas. For more information, contact us at 1-800-254-6177 or visit www.netrake.com.

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NMS COMMUNICATIONS

NMS Communications designs, delivers, and supports technology-leading systems and system building blocks for innovative voice, video, and data services on wireless and wireline networks. NMS products and services help the world's top communications equipment suppliers, solution developers, wireless and wireline operators bring their applications and services to market faster and at lower costs.

Headquartered in Framingham, Massachusetts, and with offices around the world, NMS has a rich history of technology innovations that advance the growth of the global communications industry. These achievements include the only voice quality enhancement software pairing noise reduction and compensation to

create an optimal listening environment for both parties in a mobile phone conversation, the world's first complete open IP gateway, the first complete open packet media server, and the first complete open solution for multimodal voice and data applications and services.

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NORTEL

Nortel Networks is a global leader in networking and communications solutions and infrastructure for service providers and corporations. The Company is at the forefront of transforming how the world communicates, exchanges information and profits from the high-performance Internet through capabilities spanning Optical Long Haul Networks, Wireless Networks and Metro Networks. Nortel Networks does business in more than 150 countries and can be found on the Web at www.nortelnetworks.com.

Nortel Networks has fully embraced the Session Initiation Protocol (SIP) for delivering next-generation multimedia communication services.

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QUINTUM

Quintum has developed a complete line of VoIP switching solutions that are deployed in enterprise and service provider networks around the world. The Quintum Tenor® VoIP solution is the only product that has a unique MultiPath architecture that supports transparent deployment in existing voice networks. The Tenor also supports Quintum's patented SelectNet™ switching that will monitor the QoS for each call, and in the event that call quality is threatened, it can be transparenty switched to the Public Switched Telephone Network (PSTN) - in mid call without dropping the call!! Quintum is also the only company to offer an intelligent VoIP access/switching solution, making Tenor® switches the simplest VoIP solution to deploy on the market today.

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RAD ISION

RADVISION is a leading provider of technology and products that enable real-time voice, video, and data communications over packet networks, including the Internet and other networks based on the Internet protocol or IP. RADVISION's technology and products are used by RADVISION customers to develop systems that enable enterprises and service providers to use next generation packet networks for real-time IP communications. RADVISION is actively involved in the development of the industry standards that are driving the emergence and growth of the use of packet networks for real-time communications, and was the first-to-market with enabling technology and the products required for the transmission of real-time voice, video and data over packet networks. As a

result, RADVISION is well positioned to lead the market with products and technology that enable enterprises and service providers to migrate their voice and video communications from traditional telephone networks to next generation packet networks. The company's technology and products include software toolkits, standards-based gateways and conferencing bridges.

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SIEMENS

Siemens Information and Communication Networks (ICN) is the world's number one supplier of voice and data networks for enterprises, carriers and service providers. Its comprehensive portfolio comprises IP-based convergence solutions, a full range of products for broadband access, and optical transport networks. Additionally, ICN offers comprehensive integration, services and applications support, thus providing complete solutions from a single source for the infrastructure of the Next Generation Internet. For enterprises of all sizes, Siemens HiPathâ convergence architecture strengthens business communications by enabling a company's existing voice and data infrastructures and applications to interoperate globally over all networks. www.siemensenterprise.com.

SURPASS for Building the Next Generation Network

SURPASS® provides a complete set of modular voice-data solutions for building the Next Generation Network: Voice over IP Virtual Trunking, Next Generation Local Switch, Signaling Overlay Network to bridge all types of networks, and Multimedia Applications for a clear competitive advantage. www.siemens.com/wesurpass

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Sun microsystems

Sun Microsystems, Inc. was founded with one driving vision. A vision of computers that talk to each other no matter who built them. A vision in which technology works for you, not the other way around. Sun has long been synonymous with leading edge technology. Now, after 18 years of telling the world "The Network is the Computer," Sun is poised to become the leader in the emerging network-driven economy. Forward thinking organizations are looking to Sun to lead them into the dot com future.

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Thinkengine NETWORKS INC

Thinkengine Networks, Inc. develops and markets carrier class voice services platforms specifically architected for the delivery of converged PSTN and VoIP advanced voice services, a rapidly growing market segment in the telecommunications industry. The company's scalable and highly integrated platform enables a full range of voice-activated services, while providing the lowest total cost of ownership for carriers and service providers. Thinkengine is dedicated to the application of SIP to deliver next generation enhanced services while at the same time providing a bridge from legacy PSTN networks to the VoIP future. Using Thinkengine solutions, carriers can now profitably offer speech activated enhanced services such as voice driven mid-call conferencing, voice driven voicemail, and voice driven follow-on portal services. The Thinkengine platform can be deployed in PSTN and IP networks today. Thinkengine Networks was founded in December 2000 and is headquartered in Marlborough, MA.

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Ubiquity

Ubiquity Software Corporation develops and markets SIP-based communications software for service providers, Independent Software Vendors (ISVs) and OEM partners worldwide. The company offers two products, its award-winning SIP (SIP A/S) and Speak Conference Director (SCD). Through its SIP Applications Partner Program, Ubiquity makes available for deployment a suite of third party mobile and wireline SIP-based applications powered by the Ubiquity SIP Application Server. The company has 100 and corporate offices in three countries.

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- State of the art of ip telephony protocol design -- MCI's Henry Sinnreich assesses IPDC, shows why H.323 is badly broken, and describes how hybrid IP and PSTN networks will rapidly become all IP nets

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Posted: 05/1998

The Cure for the Common WAN Service

How a simple LAN interconnection service can open the door to offering value-added data services

By Katherine Demacopolous

Volumes have been written about the importance of delivering value to customers and the necessity of responding directly to customers' needs.

But, what exactly is meant by responding to customers' needs and what makes a service offering "value-added"? Consider the case of local area network (LAN) interconnection. True, any interconnection service offering--leased lines, frame relay, native asynchronous transfer mode (ATM) service--addresses the issue of getting the customer's data from site A to site B. But the real difference becomes apparent when considering how the service is delivered and who is responsible for its various components.

With a private line solution, the management information systems (MIS) manager is responsible for creating the architecture of the network, managing the terminating equipment and troubleshooting. To add, move or delete a network location, equipment at each of the existing locations must be reconfigured.

With frame relay and native ATM services, the end user is responsible for testing, purchasing and installing wide area network (WAN) equipment to interface to a service provider. When the components are in place, the end user conducts end-to-end testing between network locations. If any problems arise, troubleshooting must be coordinated between all the equipment vendors involved and the service provider.

A true value-added service offering would be one in which the service

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provider takes over the management of the wide area component of the customer's network. The customer signs up for a service level agreement, and the service provider creates the network design, builds in the appropriate level of network redundancy and provides the troubleshooting and operations support. Transparent LAN service is such a service offering.

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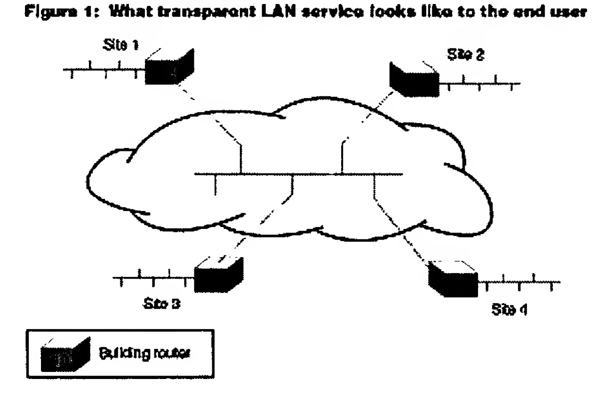
Transparent LAN Service

Simply put, transparent LAN is a high-speed LAN interconnection service that hides the complexity associated with wide area technology, design and management from the end user.

With transparent LAN service, a service provider interconnects a corporation's LANs in such a way that the wide area is transparent to the end user. All of the end user's LANs appear to be on the same LAN, regardless of where they are physically located. Transparent LAN service gives the appearance that a set of sites are all connected to the same LAN segment.

Think of transparent LAN service as the service provider building a "campus backbone" over the wide area, where each site represents a location on the campus and the transparent LAN service acts as the campus backbone LAN. The network that supports this service offering is a sophisticated, robust network, typically based on ATM, but the complexity of the wide area is hidden from the user (see figure 1).

Figure 1: What transparent LAN service looks like to the end user



The most natural model for transparent LAN is for it to operate at full LAN bandwidth (for example, 10 mega-bits per second--mbps for Ethernet). However, by using traffic management capabilities inherent in their network infrastructure, service providers also can offer transparent LAN connections at subrated speeds. For example, a native Ethernet connection could be offered at 2mbps or a Fast Ethernet connection could be offered at 30mbps.

Trials and Tribulations

To understand how transparent LAN service enables a service provider to solve its customers' business problems, let's consider some of the conflicting pressures MIS people currently are facing:

- **Great expectations.** Companies in many industries are expanding rapidly, adding new locations, acquiring other companies and partnering with organizations across the globe. Users expect real-time connectivity with individuals around the world. They anxiously anticipate the integration of an acquired company's network with the parent company's network. As soon as employees are moved into a new building, they want to be able to communicate with the rest of the company as if all employees are working under the same roof.
- **Budget crunch.** Despite the fact that companies are demanding more from their networks (and from MIS), upper management tries to keep costs to a minimum and often looks to cut MIS budgets.
- **Time demands.** With everything an MIS manager is responsible for, time is a valuable commodity. Eliminating the complexity of WAN planning, deployment and management frees MIS for other activities.
- **WAN skills shortage.** It is becoming increasingly difficult for corporations to hire and keep skilled WAN expertise. Because of the great demand, salaries for these positions are rising sharply. It is challenging for a corporation to attract skilled WAN managers in large part because these experts often can enjoy a better career path if they choose to work for a service provider.
- **Complexity of WAN technologies.** WAN technologies such as frame relay and ATM are complicated and may be unfamiliar to many MIS employees. Corporations have been reluctant to deploy ATM in LANs in large part because of the overwhelming complexity of the technology. As a whole, MIS prefer to work with familiar LAN technologies.

Benefits of Transparent LAN Service

The main benefit transparent LAN service delivers to the end user is simplicity. With transparent LAN service, the WAN is as easy to manage as a LAN. The customer does not have to learn complex technologies such as frame relay or ATM. The customer works only with familiar LAN technology and leverages the service provider's expertise in the wide area. The customer doesn't have to worry about network design, working with multiple vendors, hardware maintenance agreements, hardware or software installation, equipment configuration or testing. The customer simply hands the service provider a LAN extension from each site to be interconnected and the service provider installs, monitors, troubleshoots and otherwise manages the network.

Following are some additional benefits which demonstrate how transparent LAN service satisfies corporations' requirements:

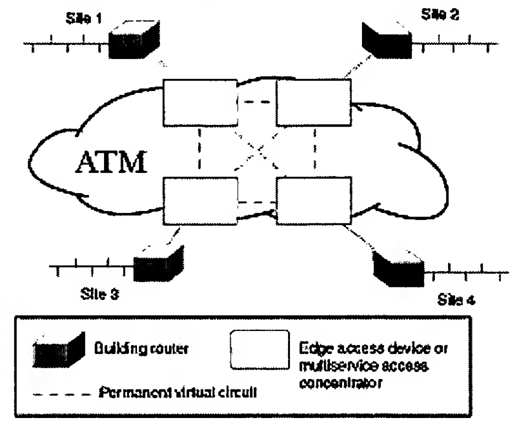
- **High speeds.** Transparent LAN service is offered at full native LAN speeds (10mbps Ethernet, 4mbps or 16mbps Token Ring, 100mbps Fast Ethernet or FDDI). Note that when the service is offered at native LAN speed, there is no speed mismatch between the LAN and WAN and, therefore, no bottleneck.
- **Just the right speed.** In the event that a customer doesn't require full LAN speed bandwidth, the service can scale to the appropriate speeds (for example, 2mbps for Ethernet LANs or 20mbps for Fast Ethernet LANs). The customer pays only for the bandwidth needed. When a customer's needs change, the service can be upgraded without requiring a change in equipment or network configuration.
- **Cost savings.** Subscribing to a transparent LAN service is far less expensive than the cost of building, managing and maintaining a WAN, especially when the costs of recruiting, hiring and training administrative and technical personnel are factored in.
- **Less risk.** Making the wrong technology decision can negatively impact productivity and the bottom line. With transparent LAN service, companies can minimize risk and protect themselves from technology obsolescence. Transparent LAN service protects users from having to reinvest in WAN customer premise equipment (CPE) with every technological enhancement or upgrade. Because the service provider extends its coverage into the end user's network locations through a LAN interface, the end user's investments can be centered on keeping the LAN and corporate applications at the leading edge.
- **Efficient utilization of resources.** With transparent LAN service, companies can centralize their server resources, since employees at each site enjoy high-speed access to servers housed in a single location. Centralizing resources allows companies to enjoy cost savings because of the reduction in capital equipment and maintenance costs.
- **Focus on core competencies.** Corporations want to get out of the business of building networks and focus their personnel and financial resources instead on what they do best--be it banking, healthcare or making jeans. By outsourcing networking function to its service provider, a company can channel capital, management and personnel resources into the company's core business.
- **Future-proof solution.** With transparent LAN service, it is easy to add new sites and upgrade to higher speed service as needs change. And because the customers did not purchase edge equipment or invest in a particular wide area technology, they are not at risk if the wide area technology of choice changes.

Implementing Transparent LAN Service

The network that best supports transparent LAN service has an ATM infrastructure surrounded by a tier of edge devices or multiservice access concentrators. These edge devices can be located either in a service

provider's point of presence or at the customer site. The service provider will connect from the edge device to the customer's LAN segment at each of the sites to be interconnected. The edge devices are then interconnected via permanent virtual circuits (PVCs), either in a full mesh or, in some cases, in a more streamlined topology. With the appropriate separation of customers' traffic both in the edge devices and in the allocation of PVCs, the service provider has created a virtual private network for that individual customer (see figure 2).

Figure 2: How transparent LAN service is implemented



Transparent LAN service can address many problematic scenarios:

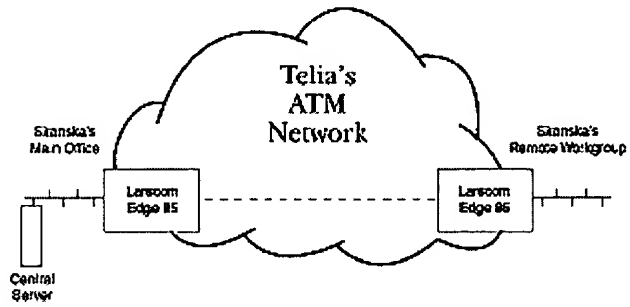
- How can I provide all of my regional offices with high-speed access to centrally located servers?
- How can I continue to run my business while half of my employees are changing locations?
- How can I ensure the information my employees need to access is always available to them--even in the case of a natural disaster?

Telia, the largest network service provider in Sweden, used transparent LAN service to solve a problem for one of its customers.

Skanska, a large construction firm in Sweden, wanted to relocate a workgroup from the company's headquarters so it would be closer to a significant customer. A LAN was installed for the workgroup in the new location. The staff needed access to the central resources at Skanska's main location, but Skanska did not want to install a local server and it wanted to retain the concentration of server capacity at Skanska's main facility. The network linking the remote group and the main building mandated sufficient capacity to support the rapid and efficient transmission of large volumes of data, such as technical drawings.

Skanska subscribed to Telia's transparent LAN service offering at native Ethernet speed, which provided Skanska with a transmission capacity of 10mbps (see figure 3). The link provided by Telia was completely transparent to Skanska.

Figure 3: Relocation service application of transparent LAN service. Providing a remote workgroup with 10mbps access to centrally located servers.



Because it subscribes to the transparent LAN service, Skanska is able to maintain its servers and the technical personnel who manage those servers at a single location. Also, with this solution, all network layer addressing is preserved. Network administrators are not forced to deal with the creation of new subnets and the resulting painful reconfiguration of user workstation addresses. And the remote workgroup is able to access all applications and data from the centrally located servers as quickly and easily as if the employees were located down the hall from those servers.

A First Step into Value-Added Services

Transparent LAN service is one way service providers can start taking a "value-added" approach to offering services. And, transparent LAN service can serve as a platform for the service provider to offer additional value-added services over the same infrastructure. Because the service provider connects directly to the customer's LAN, as opposed to a foreign wide area interface, the service provider can provide routed connections to the Internet, secure community of interest networks, information sources and more, all from the one LAN connection point. In other words, once a service provider has a transparent LAN service customer, the service provider can offer additional value-added services, such as high-speed Internet access and intranet or extranet services over the same connection to the customer's LANs.

Offering value-added services can be the keystone of a competitive strategy for CLECs. With transparent LAN service, for instance, the service provider offers much more than just high-speed pipes. The service provider reduces complexity, risk and cost of ownership. End users receive an affordable and easy solution. Service providers have an opportunity to offer true value to their customers by not only solving their internetworking problems, but by helping them address their real-world business problems as well.

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Last updated 02/10/2005 14:30:08 by Henning Schulzrinne

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Hold the (video) phone

ITU looks to add new features to H.323 videoconferencing spec.

By Sandra Gittlen
Network World Fusion, 04/14/99

■ BREAKING NEWS

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LOS ANGELES -- If you can put phone callers on hold, why not videoconferencers?

Call holding and message waiting are among the new features being added to the International Telecommunication Union's H.323 videoconferencing standard.

Jay Gilbert, a technical engineer at Intel's Architecture Labs in Hillsboro, Ore., told attendees of his Spring Internet World '99 session that H.323, an effort of the International Telecommunications Union, could see a third revision as early as this month.

A key goal is to make H.323 products easier to use and more efficient, Gilbert said.

He acknowledged H.323 has been slow to take off, in part because of concerns over its complexity, despite the efforts of companies such as Intel, which sell H.323 gear. Some companies are now looking to add multimedia capabilities based on the Session Initiation Protocol, a proposed standard by the Internet

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Intel Architecture Labs H.323 page

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White Pine jumps on H.323 bandwagon
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H.323, which was adopted as a standard in 1995, defines packet-based communication over digital transports. Originally designed for LANs, H.323 has gone through several changes to make it useful over WANs and the Internet.

In addition to the phone-like features, the new version will add a directory system that lets users look each other up by name instead of by numerical IP address and the ability to track connections for billing purposes. Gilbert said it would also use less server memory.

Gilbert said such features would bring H.323 closer to the functionality of the traditional phone system.

Intel Architecture Labs, a research arm of the microprocessor company, co-wrote several parts of H.323 and has pushed for its widespread adoption. The Intel Video Phone is based on the standard, as are other Intel multimedia products and research efforts.

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Posted: 04/1999

Integrated Access Devices

Give ATM New Life in Local Loop

But Good Old TDM Assures Voice Quality and CLASS Features

By Peter Lambert

In 1999, industry eyes are turning to what many see as the last bottleneck of inefficiency in public network architecture: last-mile access.

The solution to that inefficiency, some say, lies with asynchronous transfer mode (ATM) cell switching. It was designed from the start to answer the now-urgent service provider demand for equipment that integrates voice, data and video over a single, broadband access connection, while also provisioning and policing quality of service (QoS) for each of those services in that single pipe.

The central pitch: ATM Layer 2 transport enables service providers to provision a virtual circuit (VC) for each traffic type, and each VC can be assigned a QoS performance level to accommodate the demands of Layer 3 services such as data virtual private network (VPN) or delay-sensitive packet voice service.

In turn, QoS mechanisms enable service providers to offer specific, network performance-based service level agreements (SLAs) to their customers, a key to premium service differentiation in the bewildering clutter of competitive local eXCHANGE carriers (CLECs), incumbent LECs (ILECs) and Internet service providers (ISPs).

"CLECs can only compete with ILECs using technology that is both cheaper and more capable than traditional, fixed-bandwidth, TDM

(time-division multiplexing) technology," says Tom Barsi, marketing vice president for VINA Technologies Inc., Fremont, Calif. VINA supplies ATM-based Multiservice XCHANGE integrated access devices (IADs) to Tampa, Fla.-based CLEC 2nd Century Communications Inc. "It's natural for CLECs to go to packet voice, because a 40 [percent] to 50 percent cost reduction will lower their entry barrier by eliminating TDM cross-connects and Class 5 switches, and gain dynamic bandwidth in the access segment," he adds.

In 2nd Century's case, VINA's access device will work with Boca Raton, Fla.-based Siemens Communication and Information Networks' ATM switches to deliver data VCs to a customer's ISP and voice VCs to a customer's long distance telephone carrier.

Naysayers to this vision doubt the abilities of either ATM transport of packets or direct Layer 3 Internet protocol (IP) packet switching to support the call-control features that customers expect from their TDM, Class 5 and 4 voice switches--features ranging from call waiting to private branch eXCHANGE (PBX) to 800-number translation.

Yet ATM IAD makers such as VINA say they'll bring increasing support for enhanced Centrex, unified messaging and other voice applications into their packet- and cell-based products this year. Similarly, Sonoma Systems Inc., Marina del Rey, Calif., intends to deliver circuit emulation software for voice support to its ATM platform "soon." This could enable it to pass packet voice traffic through gateways and TR-303 and TR-008 interfaces to Class voice switches.

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The CLEC Business Model		
	Traditional CLEC Model	Next-generation CLEC model
Initial Investment per city	$7 million	$700,000
Capital Investment	$800-$1,200 per line served	<$500 per line served
Revenue Requirement	$80 per line served	$40 per line served
Projected Local/LD Revenues	$60 per line served	$60 per line served
Net Profit (loss)	-$20 per line served	$20 per line served*
*Plus revenue from value-added services		

IAD Model

To fuel service-provider differentiation, dozens of new and established integrated access equipment suppliers are offering lightning in a pair of bottles: one, designed as customer premises equipment (CPE), is the IAD; the second, sitting at the central office (CO) or point of presence (POP), is the multiservice access (MSA) device.

In general, the IAD includes ports for phones, PBXs, routers and

modems--a single box integrating all traffic over one, rather than many, access lines. Fed by the IAD, the MSA grooms each data or voice service type out of the aggregate flow, combines it with other like traffic, and passes it to the appropriate backbone provider. All agree: This architecture reduces cost by "collapsing" multiple physical lines into one physical line, and by unifying network management onto a single platform.

Combined with advancing high-capacity, local-loop technologies such as digital subscriber line (DSL), the IAD-MSA architecture promises to make access networks more future-proof.

"There's a big opportunity for high-bandwidth community and aggregation of high-bandwidth traffic," says John Stormer, vice president of marketing for national DSL CLEC NorthPoint Communications Inc., San Francisco. "The application side is just beginning to realize the possibilities of broadband."

Although NorthPoint has "no intention of ever owning a Class 5" voice switch, it could pass packet-voice traffic to either circuit-voice or packet-voice carriers and is now "in the thick of developing" integrated data and voice transport over DSL, Stormer says. "There are both cost and performance issues, such as how to handle voice or video without a way to negotiate my way through the Internet. That is a QoS issue."

Adding VC-based transport to broadband integrated access promises to enable not only QoS, but also "oversubscription" of bandwidth by dynamically allocating capacity to services as needed, rather than preconfiguring permanent TDM circuits, the bandwidth of which lies unused and wasted when phones or modems are inactive.

"TDM is very rigid, requiring that you actually touch wires and ports and devices to turn up a new service," says Kevin Walsh, marketing vice president for Accelerated Networks Inc., Moore Park, Calif., another maker of VC-based integrated access equipment for service providers and customers. "You need something that virtualizes the network, which means using ATM VCs or IP flows to turn off a VC to one ISP and turn on a VC to another ISP with a single software command."

To offer bona fide voice services, the IAD will have to become mighty complex, incorporating "Layer 2 VCs, Layer 3 routing and bridging, software PBX and voice interfaces," Walsh says. "We will have a voice over IP (VoIP) gateway connection to the Class switch, but it's not here yet for a pragmatic reason: 99 percent of voice calls remain circuit-switched."

Delivering Voice and Data Services over End-to-End ATM

Virgin Market

For now, the window of opportunity for VC-based integrated access lies with the largely untapped small- to medium-sized business market for integrated data and voice access, particularly in the deployment of DSL services. For small businesses and consumers, DSL promises broadband, integrated access over a single copper loop.

Both regional Bell operating companies (RBOCs) and CLECs have deployed ATM and, in some cases, frame relay (which also is capable of provisioning VCs for each data or voice or video session) in their DSL access networks, all the way to the customer premises.

With their 1996 joint purchase of Plano, Texas-based Alcatel USA's asymmetric DSL (ADSL) equipment, BellSouth Corp. and SBC Communications Inc. adopted ATM transport from modem to CO. Although "always-on," high-speed data plus is driving ADSL so far, Alcatel will add "voice-over-X" capabilities to its ADSL system by the third quarter of this year, says Steve Makgill, the company's director of DSL products. "Our whole ADSL infrastructure is ATM, so voice directly over ATM is a simple matter, and we can carry IP traffic, so voice over IP over ATM is not a problem for us either."

Further, last year, long distance carriers AT&T Corp. and Sprint Corp. also endorsed multiservice ATM access, and DSL CLECs Covad Communications Co. (an AT&T partner), Santa Clara, Calif.; Rhythms NetConnections Inc. (an MCI WorldCom Inc. partner), Englewood, Colo.; and NorthPoint adopted ADSL and symmetric DSL (SDSL) that employ either ATM or frame relay transport.

To advance the cause of integrated access over DSL by the second quarter of this year, DSL router makers including FlowPoint Corp., Los Gatos, Calif., and Netopia Inc., Alameda, Calif., will combine forces with packet-voice gateway makers CopperCom Inc., Santa Clara, Calif., and Jetstream Communications Inc., San Jose, Calif., to create IADs that will accommodate up to 16 toll-quality voice VCs, along with a high-speed data VC, over single copper pair. Thanks to ATM statistical multiplexing, whenever any of those virtual voice circuits is unused, the bandwidth is reallocated to the data VC.

With 1.2 million high-speed DSL (xDSL) lines installed worldwide, PairGain Technologies Inc., Tustin, Calif., has developed the ATM-based Avidia System IAD and MSA. "Our IAD follows the model of putting more intelligence at the customer premises--for PBX, voice management, data aggregation, routing--setting up VCs from CPE to CO," says Kevin Woods, director of Megabit Access product marketing at PairGain.

"Installing dumb DSL access multiplexers is expensive and won't handle large-scale IP voice and other advanced services," Woods says. "Why

put up a permanent 64-kilobit analog voice circuit when quality voice can be delivered with about 8- to 10-kilobit ATM variable bit-rate VCs and some compression?"

For larger businesses and multiple-tenant units unable to justify the cost of 45 megabits-per-second (mbps) T3 optical network access, IAD vendors including ADC Telecommunications Inc., Minneapolis; Sentient Networks Inc., Milpitas, Calif.; Sonoma and VINA also expect to deliver inverse multiplexing over ATM (IMA), which can "bond" multiple 1.5mbps, T1 access lines together as if they were one, broadband pipe.

For example, the Sonoma Access multiservice ATM access device, combined with Vienna, Va.-based Advanced Switching Communications Inc.'s Multiservice Aggregator, is optimized for multiplexed services over three to four bonded T1s. Shipping since last June at less than $10,000, Sonoma Access supports Ethernet, fast Ethernet, token ring, dial-up modem and other customer interfaces, while delivering T1, IMA, T3 or OC-3 (155mbps) access to the carrier.

"All this allows service providers to offer new, profitable services--grooming voice from data, Internet access, telemedicine, distance learning--that you couldn't do at high-access facility price points," says John Mazzaferro, Sonoma's marketing vice president. "The key is oversubscription to save on facilities costs and QoS. Service providers are offering SLAs, and they can't do that without QoS."

Not So Fast

By pulling CLASS (customer local area signaling service) voice feature software off of Class 4 and 5 switch hardware and storing it in call-control servers, ATM integrated services would use the servers to apply features to packet-voice traffic, then pass the traffic directly to packet backbones or through gateways to the public switched telephone network (PSTN).

However, many industry players question the speed with which robust call-feature sets can be developed for the call-control server model.

"With voice, you need to be rich with features, most of which are generated by Class 5 and Class 4 switches," says John Marble, vice president of Telco Systems Inc., Norwood, Mass. Telco Systems' EdgeLink 300 IAD supports Internet and transparent local area network (LAN) services (TLS) via packet transport, but it also supports up to 30 voice circuits via standard TDM.

"Packet-to-circuit gateway makers are seeking to replace Class 4 switches with packet switches, where the port volumes are highest, but it will take a long time to replicate all Class 5 features on the packet side," Marble says. "Initially, like any technology, packet voice will be feature-

poor, which is where carriers make all their money. And while packet in the long term may prove most efficient, there are a lot of interim steps first to protect that TDM feature investment."

According to frame relay equipment makers, the better near-term solution is to support both TDM voice circuits and packet data over integrated access links. That way, voice features are maintained. At the same time, frame relay vendors including Adtran Inc., Huntsville, Ala.; Paradyne Corp., Largo, Fla.; and Sync Research, Irvine, Calif., are adding to their equipment "probes" designed to report frame relay permanent VC (PVC) performance and so to support data SLAs.

"It could be 10 TDM circuits dedicated to the PBX and six DS-0s that look like one pipe with one or 100 PVCs for data," says Scott Eudy, vice president of Paradyne's network access division. "In the past, packet data over frame could be sold as riding for free over the extra DS-0s in your voice network, and now with data growth, the message is analog voice rides for free on your data network."

Consequently, ADC; Carrier Access Corp., Boulder, Colo.; Paradyne; Premisys Communications Inc., Fremont, Calif.; Telco Systems and other access vendors continue to offer a mix of packet and circuit transport over an integrated access link, as do frame relay-based DSL equipment makers, including Fujitsu Network Communications Inc., Richardson, Texas.

Still, some of those vendors say their equipment will accommodate migration to packetized and/or cell-based integrated access. Currently, for example, Paradyne's SuperLine IAD combines dedicated TDM circuits with frame relay data transport to the CO, but "well before the end of the year, we'll unveil multiline voice over IP-over-DSL," says Ron Stein, DSL marketing director.

The leading makers of Class switches themselves may help such efforts. In February, Lucent Technologies Inc., Murray Hill, N.J., delivered its AnyMedia MultiService Module packet-switching enhancement software to ICG Communi-cations Inc.'s Class 5 switches. The module combines remote access concentrators with ATM switching to terminate either TDM or packet traffic on any Lucent 5ESS switch.

Also in February, SBC Communications began beta testing Richardson, Texas-based Nortel Networks' Succession Network, a complex of distributed ATM switches, multiservice gateways and call-control servers.

"Carriers want to cut costs by collapsing Class 5 and 4 into one layer, and they're realizing that, if you want to offer voice services, you have to have ATM, because you have to offer the same quality of service that enterprises get today," says Graham Rance, Nortel's vice president. "It's no longer a nodal system; it's the distribution of ATM switching fabric

for the purposes of distributing both access and call control."

Nortel and Lucent enjoy at least a slight edge over startup players in maintaining those TDM call-control services in the packet world. "Competitors will have to create the services," Rance notes. "We can just move them over from our switches."

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Internet Telephony System Created at Columbia Is Tested Successfully by Telecommunications Firms (photo of Prof. Henning Schulzrinne)

BY BOB NELSON

In a landmark event held at Columbia University this month, leading telecommunications firms and research organizations successfully tested Internet telephony products that promise an array of new telecommunications services -- such as a "universal address" for telephone, fax, e-mail and other services -- at sharply reduced prices.

The technology-oriented event, held April 8-9 and whimsically titled a "bake-off" by computer scientists, was organized to test the interoperability of software and hardware devices using the Session Initiation Protocol (SIP), which can set up and configure Internet telephone calls and multimedia sessions. The protocol was approved last month by the Internet Engineering Task Force (IETF), the standards body governing the technical foundations of Internet products.

Software as well as dedicated hardware passed the interoperability testing. Hardware products included Internet-to-phone gateways and "Ethernet phones" that plug directly into local area networks. The event was able to show that subscribers can move from location to location, anywhere on the Internet, with phone calls following them automatically, no matter who provided the hardware or software. Users could also forward calls to any Internet destination or a telephone number. Some groups also tested advanced features such as call screening and user authentication.

According to Internet telephony industry analyst Jeff Pulver, the event instantly advanced the state of the art in Internet-based telephone systems and services. "Internet telephony is already reshaping the global telephone system in dramatic new ways," said Mr. Pulver, who is president and chief executive officer of pulver.com, Inc., which publishes newsletters and produces trade shows related to Internet telephony. "The number and names of the companies that tested products at this event means that SIP is quickly going to have a big effect on this telecom industry overhaul."

Henning Schulzrinne, associate professor of computer science and electrical engineering at Columbia University, host and co-author of SIP, remarked that "the event showed that the SIP specification is mature and stable enough to allow a wide range of implementors to produce implementations that interoperate with minimal effort. It was exciting to see so many high-quality products based upon SIP emerge so quickly."

Internet telephony carries telephone conversations as Internet packets rather than the current digital circuits. It promises high-quality voice and multimedia, improved network efficiency, rich computer-telephony integration, advanced services, an open market for providers and reduced costs for consumers. New services include a single identifier ("universal address") for phone, cellular, fax, e-mail and paging, user control over incoming calls and easy integration among e-mail, web and telephone services.

According to a recent study by the investment bank Piper Jaffray Inc., the Internet telephony market will increase to $14.7 billion by 2003. In 1997, 70 million minutes, less than 0.1 percent of the total call volume, went over Internet-protocol networks. In four years, the report predicts, this will increase to 70 billion minutes, about 6.1 percent of all calls.

Participants at the interoperability event linked their products together to test that they will work with one another across the Internet. At the end of the event, nearly all implementations had achieved interoperability for call setup and media capability negotiation for multimedia calls. Several were interoperable on the first try, and most others after minor changes or bug fixes were made.Organizations participating in the interoperability testing came from Canada, Finland, Sweden, the United Kingdom and the United States: 3Com Corp., Alcatel, British Telecom, Cisco Systems Inc., Columbia University, Dialogic, dynamicsoft, Ellemtel, Ericsson, Helsinki University of Technology, Hewlett-Packard Co., Lucent Technologies, MCI Worldcom, Mediatrix Peripherals Inc., Nortel Networks and Pingtel. The group plans additional interoperability tests of other advanced features offered by SIP in August.

More information about SIP is at http://www.cs.columbia.edu/~hgs/sip. This document is available at http://www.columbia.edu/cu/pr/. Working press may receive science and technology press releases via e-mail by sending a message to opa@columbia.edu. 4.14.99 19,518 -more- -3-

April 16, 1999 (1:50 PM EDT)

Vendors Click On New Net Telephone Protocol

By Mo Krochmal, TechWeb.com

A new signaling protocol developed at Columbia University may let friends and family dial up through their e-mail addresses.

The new protocol is attracting the attention of telecommunications industry in the month since it was approved by an Internet standards body.

At it simplest, Session Initiation Protocol (SIP), a standard adopted in March by the Internet Engineering Task Force, is a set of instructions for setting up teleconferences.

It is a system that may have broad implications for Internet telephony, meshing it ever closer with traditional telephone technology. It may also give vendors a choice between it and the H.323 protocol used for multimedia applications such as videoconferencing over packet-switched networks.

"There is a lot of interest in this as a mechanism for creating new services," said Henning Schulzrinne, associate professor of computer science and electrical engineering at Columbia and one of the co-authors of the protocol.

Users of the technology would not be aware of it, as it resides as software in the telephone where it is controlled by servers.

In the space of two or three years, Schulzrinne said, SIP may allow calls to a single number to follow a person, regardless of where they are. Or it may allow one player to start a game of Quake and call a friend or two to join in. Or, even easier, a person could dial a friend via their e-mail address.

The new technology was tested for interoperability at the university in early April, with 16 companies including MCI WorldCom, British Telecom, Nortel Networks, Hewlett-Packard, 3Com, Cisco, Lucent, Ericsson, and Alcatel participating in the two-day exercise. Venders were paired off to make calls to each other, make changes as needed.

"They brought their source code and made fixes on the spot," said Schulzrinne. "All the vendors had been working on implementations -- we just brought them together."

Guido Schuster, distinguished member of 3Com's technical staff, said the new standard gives the industry a choice.

"[SIP] was developed from the beginning as an Internet protocol," he said. "H323 was developed at first for video conferencing over ISDN, then adopted for LANs and WANs. SIP is a cool technology that gets voice to fit naturally on the Internet."

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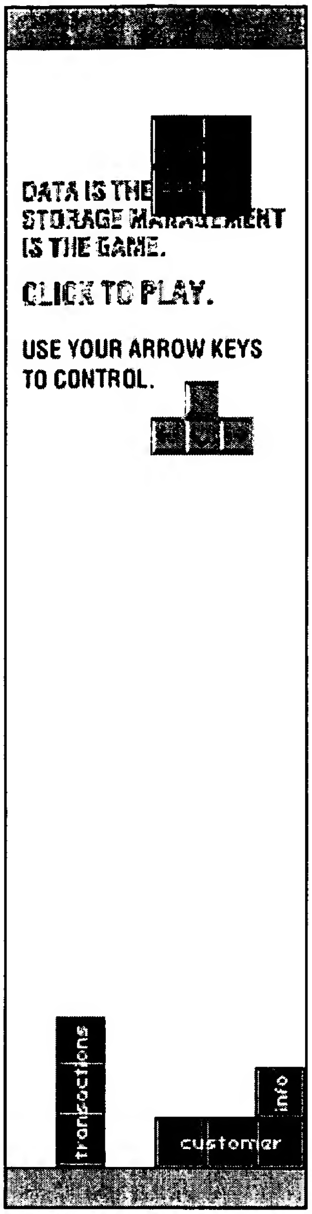
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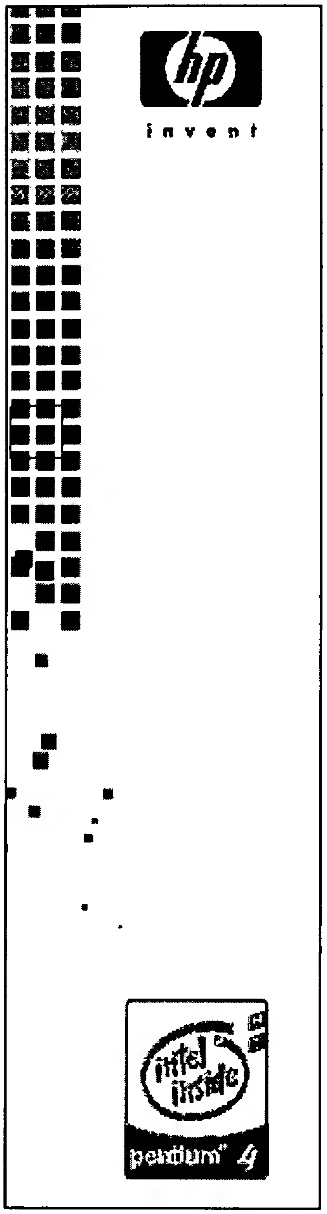
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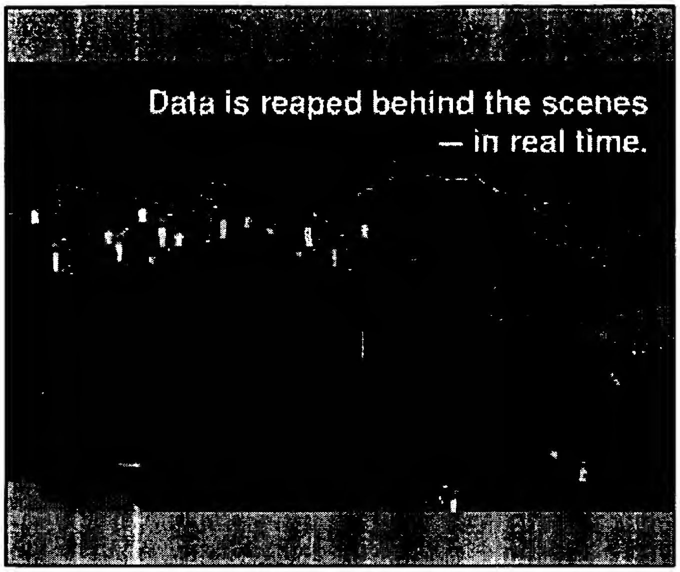
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GOT THE URGE TO CONVERGE

TECHNOLOGY ADVANCES AND STANDARDS PROGRESS ARE PUSHING ENTERPRISE-LEVEL VOICE AND DATA CONVERGENCE EVER CLOSER TO REALITY.

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By Susan Breidenbach
Network World, 09/27/99

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The vision of a single network delivering voice and data to enterprise desktops has been like a mirage: It never gets any nearer and disappears if you look too closely. But the images of convergence may finally be more than thirst-induced fantasies.

COLUMNIST CALL: Hear what Nolle, Bradner and Kobielius say about convergence. (In G2 RealAudio)

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Thanks to technology advances and emerging standards that are enabling a new generation of products, convergence is moving beyond call centers and other niches. The traditional PBX at corporate headquarters is in no immediate danger, to be sure, but small precursors to its ultimate IP-based replacement are starting to appear in branch offices.

The original premise of convergence - to save money by collapsing two network fabrics into one - is still valid, but the focus is shifting to the applications, enhanced communications and ease of management a single network enables.

Think about how frustrating it is to repeat tracking information to successive customer service representatives when you call a courier service to check on a shipment and get transferred five times. A converged voice/data network turns voice into an

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application that can be integrated with database software so tracking information can move with the call.This means database vendors can start voice-enabling their products.

"This is the sort of thing that will drive the migration to converged networks," says Jim Daugherty, general manager of data services marketing for AT&T.

Such enhanced communications are increasingly critical in an Internet economy in which the customer is king and your competitors are only a click away. Also important is the ability to support a legion of mobile workers and telecommuters.

"When my wife calls to tell me she's going into labor, it's no big deal for an IP-based switch to look into my scheduler, find the conference room I'm in and forward the call there," says Christian Renaud, manager of product marketing for Cisco's enterprise voice business. "Getting a traditional PBX to do that would take a whole fleet of Andersen consultants."

Because IP is distance-insensitive, convergence will help employers cope with tight labor markets by letting people work from almost anywhere. With a single line, you can give home-based employees LAN access, Internet access, a PBX extension, voice mail and features such as speed dial.

Call center operations are a primary beneficiary. Anyone anywhere on the enterprise network, including remote individuals attached via the Internet, can become part of a virtual call center that works off a single database of customer and product information. Call center managers will be able to create databases of incoming calls and at the end of the day send the five worst calls to the product managers, for example.

"That's the whole ball of convergence wax - the ability to do things you couldn't formerly do cost-effectively," says Don Hausman, product manager for 3Com's voice solutions group in Andover, Mass. 3Com's NBX 100, acquired through a merger with NBX this spring, is helping define a new class of network server that functions as an IP-based PBX.

3Com, like NBX before it, has concentrated on providing a complete plug-and-play voice and data

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product to small offices that don't have resident IT staffs. While the target has been small businesses, large enterprises are starting to adopt iPBXs for branch offices.

Cisco has a competing product, acquired through its merger with Selsius Systems in October 1998. The system, aimed more at the enterprise than 3Com's NBX 100, is assembled from components that include CallManager call-routing software for Windows NT; two models of IP phones; and assorted gateway interface cards and routers needed for WAN connections. Cisco is now targeting smaller businesses with Media Convergence Server, a bundled version preinstalled in a server.

Cisco stays away from the iPBX label. "That's like calling automobiles horseless carriages," Renaud says. Convergence is moving voice onto data networks, and "iPBX" implies the opposite, he explains.

Vertical Networks, a start-up in Sunnyvale, Calif., is challenging the two data communications giants. It offers a more completely integrated box, called InstantOffice, that incorporates hub and router functions as well as IP telephony. However, Vertical doesn't deliver voice over IP to the desktop. Instead, it uses two wires - one to the PC and one to a traditional phone - from its box, says Reed Henry, Vertical co-founder and vice president of marketing.

That approach is fine with Laptop Lane, which is using InstantOffice to provision T-1 lines to the business centers it's creating in major domestic airports.

Photo: Kathleen King

"We could set up these centers without convergence, but we'd have to charge customers more and we'd make less," says Bruce Merrell, president and CEO of Laptop Lane.

Moore's Law marches on

In their crafting of more fully integrated wares, iPBX developers are benefiting from faster and less expensive processors, better coder/decoders, single-chip Display Systems Protocols and standards stabilization.

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"There's been more combination of previously stand-alone devices - router, hub, switch, PBX, DSU/CSU - into a single platform," says Tom Jenkins, a senior consultant with TeleChoice in Owasso, Okla. You don't get best of breed, but the integrated device is more reliable and easier to manage. Jenkins also sees more interoperability, due primarily to partnerships among equipment vendors.

The initial wave of iPBX platforms - including Lucent's first iPBX, scheduled for release before year-end - is aimed at small offices with less than 100 users. Several of the traditional voice PBX manufacturers have scheduled enterprise-class iPBX systems for release in 2001, and Nortel Networks has promised one by mid-2000.

Evolution vs. revolution

The question is, just how complete and scalable will any first-generation enterprise iPBX be?

Traditional PBXs have hundreds or even thousands of call-control features that have to be migrated from the highly disciplined and connection-oriented voice world to the anarchic and connectionless environment of data communications. How do you place a call on hold when it is just a bunch of packets that are mixed in with a lot of other packets?

Lucent expects traditional PBXs to host a lot of these call-control functions as iPBXs are adopted to support advanced applications that integrate voice and data.

"Most IP PBX vendors haven't even thought about doing things like call-detail recording or time-of-day routing," says Karyn Mashima, chief technology officer and vice president of strategy for Lucent's business communications group. "And scalability is always very hard to do in data networks, while voice PBXs are proven to scale to tens of thousands."

Lucent is adding IP trunks and features to its Definity voice switches so they can act as central servers and continue to handle much of the call control. Central and remote PBXs can communicate via IP, eliminating the need for costly T-1 lines. "Within a few months, we will be able to support IP phones off Definity," Mashima says.

Customers get a lot of the benefits of convergence while avoiding a forklift upgrade and retaining the traditional PBX's reliability and scalability. Of course, this approach leaves separate voice and data networks running to the desktop, supported by two different types of switches that have to be maintained and managed.

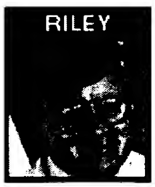

Photo: Walter Calahan

Donald Riley, chief information officer at the University of Maryland (UM) in College Park, doesn't mind that approach. It has about 120 users experimenting with Lucent's IP-enabled Definity switch and MMCX multimedia platform, as well as Cisco IP phones and 3Com wireless-IP Palms.

"What we don't understand yet is the scalability issues," Riley says. "We have 32,000 students and 10,000-plus faculty and staff, and we're not far enough into it to know if [the technology] will scale up to that level. A whole food chain has to buy into this IP telephony to make it work end to end."

Start small

Convergence may not be ready for prime time on a large scale, but you shouldn't ignore it.

"You have to start looking at it now," says David Dines, a senior analyst with Aberdeen Group, a market research firm in Boston. "Take a small branch office with an old phone system and turn it into a pilot site. Use the site to try to solve a business problem that you can't resolve cost-effectively with the old system, like using skills-based routing to send customers to the best available person in a distributed call center. See how it works and impacts the network."

Companies should also look for "green-field" opportunities. One such example would be when a branch operation is being moved into new construction office space that has no voice or data network.

"Now that the traditional PBX companies are getting into this business, there is no problem finding a reliable supplier," says Jack Chase, director of enterprise products for Natural Microsystems, a

Framingham, Mass., company that makes boards and software used in convergence products.

But make sure the equipment supports downloadable upgrades so you can accommodate new or evolving standards. The LAN side reached a watershed this past year with the ratification of the IEEE 802.1p and 802.1Q standards for QoS, and the list of products that support the International Telecommunication Union's H.323 specification continues to grow.

But H.323 was originally designed to deliver video over a single LAN and does not scale to enterprise or carrier backbones well. A better bet for a general call-control standard is the Megaco/H.248 specification being worked on jointly by the IETF and ITU. Developers have not scheduled a completion date for Megaco; nonetheless, Megaco does appear to be supplanting the IETF's proposed Media Gateway Control Protocol standard.

Anyone looking to converge large voice and data networks also should ensure that the equipment they buy can be upgraded to support Session Initiation Protocol, a proposed IETF call-signaling standard.

Soft savings

Don't expect the iPBXs to cost less than the traditional devices they are replacing. While analysts expect the prices to drop dramatically over the next year, you may pay a bit of a premium today. For example, one user says he spent about 15% more when he decided to go with 3Com's NBX 100 instead of a Toshiba key system.

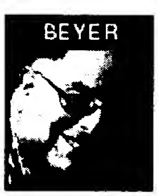

Photo: Steve Skoll

"Upfront cost wasn't the issue," says Barry Beyer, president of Disbrow Manufacturing, a producer of paperboard boxes in East Orange, N.J. He was looking for additional features, such as caller-selectable call-forwarding options and better caller-ID integration with his company's accounting system. Since March, Disbrow has been based on a converged network that uses Ethernet to deliver data and voice services to the company's 18 employees.

"The voice quality is better than what we had before," says Beyer, who was replacing a Comdial

key system. "Also, upgrades are free and can be downloaded, so I expect this platform to last a lot longer than any traditional phone system."

By consolidating administration and management, converged networks reduce operational costs, and they also make it a lot less expensive to move phones when employees change offices. Moving a regular phone attached to a traditional PBX is labor-intensive and expensive - it costs $250 on average.

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In contrast, IP phones declare themselves using their Ethernet media access control addresses - whenever and wherever they are plugged in to the network. And because the same jack is used for voice or data, network administrators don't have to tag and keep track of different types of outlets.

On the negative side, the prices of IP phones currently start at $350 to $400, while the least expensive traditional phones are in the $100 range. Also, IP phones will double the number of IP addresses that an enterprise is using, and they are subject to Ethernet distance limitations.

"That last 100 feet to the desktop is going to be the hardest to change," says William MacDonald, vice president of business development for Calista, an IP telephony start-up in San Jose recently bought by Cisco. To ease migration, Calista has come up with an interim solution: IP-based PBX extenders that IP-enable traditional digital PBXs and their telephones and turn remotely located phones into virtual extensions.

Calista reseller KLF, a telecom interconnect supplier in Ft. Wayne, Ind., is using the PBX extenders to set up virtual call centers for customers. Using a single ISDN line, KLF can provision a call center agent's home office with a phone connection, Internet link and dedicated fax line. Calls can be routed to the agent's phone just as if the phone were sitting on a desktop at the corporate site.

Five nines or bust

Reliability is a major concern as voice gets moved onto data networks, although part of this is the natural uncertainty that accompanies any paradigm shift.

But one clear danger posed by convergence is the loss of the redundancy that two separate networks provide. People who design voice switches think of downtime in terms of seconds per year because the networks have to be up at least 99.999% of the time.

In contrast, the most highly available routers on data networks today have average annual downtimes measured in minutes, or even hours. While the connectivity demands of the Internet economy are rapidly reducing the tolerance of such outages on the data side, there is still a huge difference between the voice and data requirements.

This gap must be closed because it is clear that users aren't willing to sacrifice any reliability on the voice side. When asked if they'd settle for four nines - 99.99% uptime - on a converged voice/data network that provided a lot more features and capabilities for a lot less money, the answer was a resounding and almost unanimous, "No."

"There's no reason that we can't achieve the same level of reliability in the data world, but we will never get there if we have to keep putting all these resources into evolving a separate voice network," UM's Riley says. "If we try to keep these parallel investments going much longer, the underdeveloped world could leapfrog us to convergence."

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Breidenbach is a freelance technology journalist and consultant. She can be reached at sbreide@aol.com.

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WAN Systems & Telephony WORKSHOP

Translating Menus at the VoIP Café

December 27, 1999
By Darrin Woods

Just as you raise the last spoonful of VoIP (voice over IP) alphabet soup to your mouth, the cafeteria lady appears with a huge pot, a ladle and an evil grin. "Here, dear, you look hungry; have some more," she says, as she slowly refills your bowl.

Now your spoon twirls up enough letters to spell SIP, SIP+, MGCP, H.GCP and Megaco, as well as the H.323 you've been trying to hold down for months.

H.323 was the future for organizations that had been using proprietary technology to set up and route calls until a standard emerged. Created around an existing set of protocols--including H.225.0 for setup, ISDN Q.931 for signaling, H.245 for negotiation and H.450 for additional services--H.323 provides a framework for products that interoperate over a network. But it has drawbacks.

Because H.323 is based on ISDN standards, it's limited to point-to-point only. This means simple telephony services, such as conferencing, call forwarding and call transfer, are impossible to implement without external help in the form of an MCU (multipoint control unit), which must sit in on the call and manage the sessions. For conference calls, a point-to-point session is created from the call originator to the MCU. The MCU then creates one point-to-point session from itself to each of the endpoints in the conference. The MCU must operate in a stateful mode, keeping up with all call sessions.

The need to maintain the connections makes large conference calls difficult and requires lots of horsepower. Newer H.323 implementations change this by letting cascading MCUs distribute the workload. Also, considering that they only provide point-to-multipoint or reroute servers, MCUs are expensive; consider your need for them carefully. If you can hold out for H.323 version 3, it will operate on top of UDP and run in a stateless mode, eliminating the need for MCUs.

H.323 also is a difficult standard to implement because it uses so many different protocols. Although the H.323 standard uses all binary information, several steps must be completed before compressed voice data can move back and forth, and portions of the protocol must complete their individual setups before passing control to the next sequence in the call. This can translate into six or seven round trips of information--just to set up the call.

H.323 is effective in small enterprises that need point-to-point communications between offices. Setup can be simple with phones hung directly off an H.323-enabled router, allowing office-to-office calls. Or phones can be tied in through the office PBX to the outside.

A Long, Cool SIP

SIP (Session Initiation Protocol) is much easier to use, set up and debug than H.323. Created at Columbia University, SIP is still a draft specification (RFC 2543) from the IETF, but it is finding its way into several vendors' offerings. SIP owes its ease of use to the fact that it's clear-text-based. A SIP session appears very similar to an HTTP session because both are based on similar text commands and headers. Initially, call setup would seem to take longer with SIP than with H.323. But call setup is actually quicker with SIP because it's just one modular protocol as opposed to the several individual protocols of H.323. With SIP, call setup is reduced to 1.5 round trips.

As with HTTP, unknown header information is skipped and ignored. Any required services are indicated by use of the REQUIRE header. If a required service is unavailable to a server, the server returns an error code and states the feature(s) that it does not support. The client can fall back and retry the connection with simpler features. A SIP implementation need understand only four headers (TO, FROM, CALL-ID and CSEQ) and three request types (INVITE, ACK and BYE).

SIP offers extensibility by learning lessons from HTTP as well. The protocol can be enhanced by anyone creating features or services that can be registered with IANA (Internet Assigned Number Authority, at www.isi.edu/cgi-bin/iana/ mediatypes.pl). Even small developers with new codecs or media types can have a significant impact on the protocol and its end users.

SIP also is backward- compatible, unlike H.323, which has backward-compatibility issues. With SIP, as new features are added, older ones are phased out, keeping SIP as clean and small as possible. With H.323, new features are added on top of old features that remain for the backward compatibility, resulting in further bloating of an already large specification.

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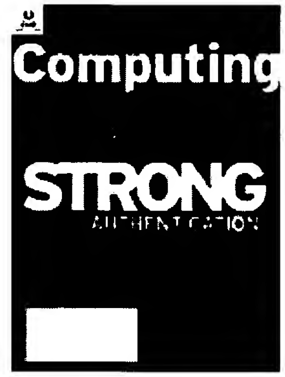
Send your comments on this article to Darrin Woods at dwoods@nwc.com.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Translating Menus at the VoIP Café

December 27, 1999
By Darrin Woods

Just as you raise the last spoonful of VoIP (voice over IP) alphabet soup to your mouth, the cafeteria lady appears with a huge pot, a ladle and an evil grin. "Here, dear, you look hungry; have some more," she says, as she slowly refills your bowl.

Now your spoon twirls up enough letters to spell SIP, SIP+, MGCP, H.GCP and Megaco, as well as the H.323 you've been trying to hold down for months.

H.323 was the future for organizations that had been using proprietary technology to set up and route calls until a standard emerged. Created around an existing set of protocols--including H.225.0 for setup, ISDN Q.931 for signaling, H.245 for negotiation and H.450 for additional services--H.323 provides a framework for products that interoperate over a network. But it has drawbacks.

Because H.323 is based on ISDN standards, it's limited to point-to-point only. This means simple telephony services, such as conferencing, call forwarding and call transfer, are impossible to implement without external help in the form of an MCU (multipoint control unit), which must sit in on the call and manage the sessions. For conference calls, a point-to-point session is created from the call originator to the MCU. The MCU then creates one point-to-point session from itself to each of the endpoints in the conference. The MCU must operate in a stateful mode, keeping up with all call sessions.

The need to maintain the connections makes large conference calls difficult and requires lots of horsepower. Newer H.323 implementations change this by letting cascading MCUs distribute the workload. Also, considering that they only provide point-to-multipoint or reroute servers, MCUs are expensive; consider your need for them carefully. If you can hold out for H.323 version 3, it will operate on top of UDP and run in a stateless mode, eliminating the need for MCUs.

H.323 also is a difficult standard to implement because it uses so many different protocols. Although the H.323 standard uses all binary information, several steps must be completed before compressed voice data can move back and forth, and portions of the protocol must complete their individual setups before passing control to the next sequence in the call. This can translate into six or seven round trips of information--just to set up the call.

H.323 is effective in small enterprises that need point-to-point communications between offices. Setup can be simple with phones hung directly off an H.323-enabled router, allowing office-to-office calls. Or phones can be tied in through the office PBX to the outside.

A Long, Cool SIP
SIP (Session Initiation Protocol) is much easier to use, set up and debug than H.323. Created at Columbia University, SIP is still a draft specification (RFC 2543) from the IETF, but it is finding its way into several vendors' offerings. SIP owes its ease of use to the fact that it's clear-text-based. A SIP session appears very similar to an HTTP session because both are based on similar text commands and headers. Initially, call setup would seem to take longer with SIP than with H.323. But call setup is actually quicker with SIP because it's just one modular protocol as opposed to the several individual protocols of H.323. With SIP, call setup is reduced to 1.5 round trips.

As with HTTP, unknown header information is skipped and ignored. Any required services are indicated by use of the REQUIRE header. If a required service is unavailable to a server, the server returns an error code and states the feature(s) that it does not support. The client can fall back and retry the connection with simpler features. A SIP implementation need understand only four headers (TO, FROM, CALL-ID and CSEQ) and three request types (INVITE, ACK and BYE).

SIP offers extensibility by learning lessons from HTTP as well. The protocol can be enhanced by anyone creating features or services that can be registered with IANA (Internet Assigned Number Authority, at www.isi.edu/cgi-bin/iana/ mediatypes.pl). Even small developers with new codecs or media types can have a significant impact on the protocol and its end users.

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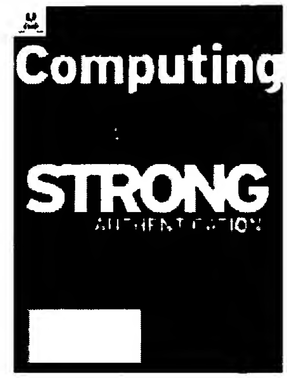
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SIP in the Press 2000

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- Exchanges: IP Service Providers in Disguise, Doug Johnson, *Sounding Board Magazine*, November 2000.
- Hey Buddy..., *Computer Telephony*, November 2000.
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- SIP Moving into Prominence for Applications, Protocol is Beyond Talking Stage to Actual Trials and, Soon, Deployments, Vendors Say. *Sounding Board*, November 2000.
- Vendors Tap Layer 7 Switching for VoIP, by Phil Hochmuth *Network World*, October 23, 2000.
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- "Web Phone Calling: Are We There Yet?", Zdnet, Anchor Desk, Berst Alert Oct. 11, 2000.
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- Rumors And Ruminations, *Internet Telephony*, October 2000. ("According to my top-secret Redmond source, the company is embarking on a wholesale change in its vision statement to a communications centric point of view -- in part driven by its recent "Dot Net" initiative. Microsoft is considering SIP-enabling the next version of NetMeeting, ...")
- Next-Generation Service Creation, *Computer Telephony*, October 2000.
- Good and evil in Atlanta, *Telephony*, Sept. 18, 2000.

  > As for softswitches, they are "being used for doing old proprietary stuff," Eriksson said. "H.323 is the last dinosaur of the telcos, and it's going to go away." Telia is expanding its intra-company trial of SIP communications from the Swedish carrier's 27,000 employees to its 660,000 ISP customers. Working with Ubiquity Software, Telia will offer a specially targeted SIP offering for teens.

- IP PBXs scale for the enterprise, *Network World Fusion*, Sept. 18, 2000.

  > We found that Version 2 of the ITU-T's H.323 standard is the most widely supported voice-over-IP standard today, with 88% of the products supporting it (see Figure 4). However, H.323 does not appear to be the long-term direction of the IP-PBX vendor community. Only 13% of the products currently claim support for H.323 Version 3. ... [T]he big trend is clearly toward Session Initiation Protocol (SIP), which was published as an Internet draft document last year. While only 13% of the products we surveyed support SIP now, more than two-thirds of the vendors said they will implement SIP over the next year.

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- Vendors SIP new promise of telephony, *InfoWorld*, Sept. 13, 2000.
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Session Initiation Protocol (SIP) & the Interaction Center Platform® Technology

Total Interaction Management from a *Single* Platform

- Why SIP?
- Why the Interaction Center Platform?
- Interaction Center Platform/SIP Benefits
- White Papers and SIP Data Sheet

Why SIP?

In the push toward Internet Protocol (IP) telephony for business communications, many approaches are available to adapt packet-based telephony methods. Among these approaches are proprietary communications protocols, point-to-point only strategies, sophisticated device identification, and carrier-class and enterprise-oriented call management/call control methods. While there are many options, however, one thing is clear - standards and simplicity are critical if the business world is to accept, and support, IP telephony.

One IP telephony approach quickly gaining support is the Session Initiation Protocol. And the reasons Interactive Intelligence now provides SIP support in our Interaction Center Platform technology are that SIP is:

- Widely regarded as the successor to H.323 for IP-based telephony
- Gaining increased attention and visibility due to major technology supporters
- An alternative to TAPI-based IP telephony models
- A protocol that eliminates the need for a separate IP-PBX and contact center solution
- The emerging standard for session control for a variety of other communications mediums
- Software-based, open and lightweight, allowing organizations of all types to support the new breed of SIP phones along with soft phones, analog phones, desktop PCs, and even mobile devices and PDAs

SIP also provides the perfect blueprint for real-time voice communications, text-based messaging and application sharing - features highlighted in Interactive Intelligence software products built on the Interaction Center Platform.

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Why the Interaction Center Platform?

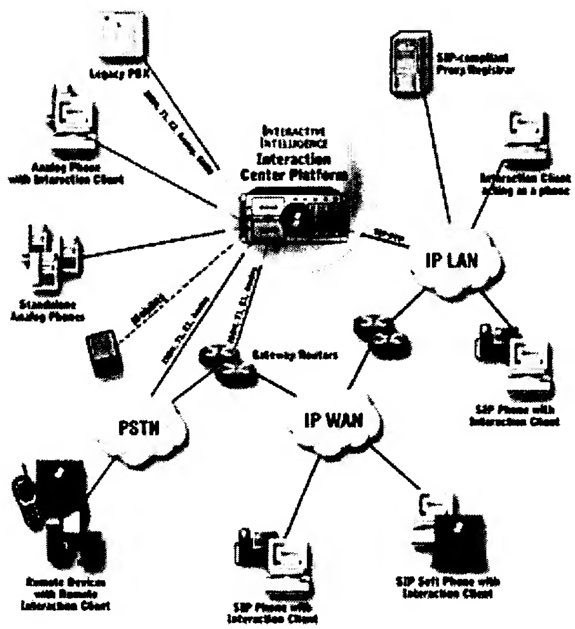
Though SIP-based soft switches are made for next generation call transport over packet

networks, they lack the interaction management applications needed for an Interactive Economy. For example, inherent features such as voice mail still require multiple communications servers and boxes within a SIP network.

Which is where the Windows® 2000-based, SIP capable Interaction Center Platform technology comes in.

In addition to replacing multi-box communications systems with a **single server**, this elegant multi-channel platform supports a full suite of interaction management software products that deliver voice mail and more. It's also the first - and thus far *only* interaction management technology - to seamlessly integrate with SIP-compliant Proxies and Registrars in addition to new IP telephony boards.

As a result, the Interaction Center Platform can drive entire SIP oriented networks from just one server while also opening the door to new SIP solutions from Voice over IP (VoIP) infrastructure providers.

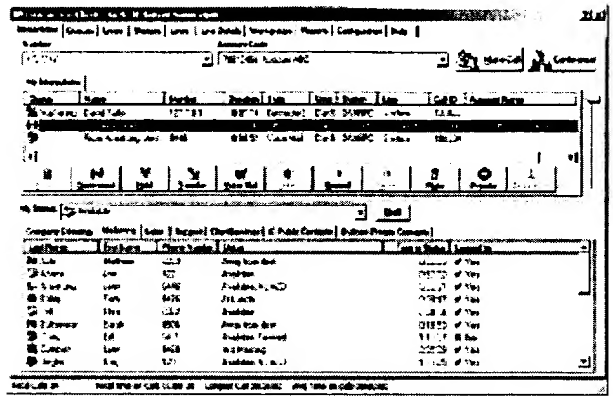


Better yet, organizations don't need to add hardware boxes to their communications systems to get advanced interaction management features. Nor do they lose such features by using SIP.

Because the Interaction Center Platform supports the SIP RFC, organizations leaning toward SIP standards can now get a full range of voice- and Web-based interaction management features via a converged voice/data network. Using IP telephony boards for media processing also provides gateway-like functionality for existing or additional ISDN, T1, E1, or analog connections to legacy PBXs and analog phones. Simply implement one of the software products built on the Interaction Center Platform within the SIP network and businesses get:

- **IVR**
- **ACD**
- **Call monitoring**
- **Voice mail**
- **Unified messaging**
- **Web-based interaction functionality**
- **Skills-based routing**
- **Screen pop**
- **CRM and ERP application integration**
- **Presence management** (Available, Out of the Office, On Vacation, etc.)
- **Comprehensive reporting and supervision**
- **E-mail and Web auto response** to frequently asked questions using e-FAQ®

- **Customer/agent Web collaboration**
- **Built-in multi-lingual support**
- **Optional wireless application interfacing with Mobilité™**
- **Remote worker/wireless user support via Communité®** for "anytime, anywhere" unified communications



Get full control over SIP calls and blended inbound/outbound interactions via the *Interaction Client®* interface

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Interaction Center Platform/SIP Benefits

Whether an organization occupies a single location or has multiple branches, IP telephony offers benefits from more efficient moves, adds and changes, as well as from centralized telecom routing configurations to distributed agents and virtual contact centers.

Thanks to the Interaction Center Platform's flexible circuit telephony and IP telephony capabilities, organizations can easily migrate to new architectures and move applications to new locations with no application rewrites. Such flexibility is especially cost-effective for organizations currently utilizing Cisco-based or SIP-oriented IP telephony, or that will consider other emerging voice technologies and architectures in the future.

Also as IP telephony cost savings move from carrier networks into the enterprise, businesses will need to move business routing and customer service applications to these new architectures. Just as with the adoption of PCs and LANs versus mainframe-style systems, IP telephony will continue to emerge as the predominant technology for communicating in the Internet Age and staying competitive in the Interactive Economy. To that end, Interactive Intelligence software products built on the SIP-compliant Interaction Center Platform provide affordable alternatives to traditional telecom solutions by offering:

- **Full media control** to perform application functions such as:
    - Playing and recording audio for IVR, ACD, auto attendant and unified messaging applications
    - Multiparty conferencing used as a conference bridge or for addressing complex customer service scenarios
    - Supervisor monitoring for ACD
    - Call analysis and call progress detection for outbound predictive dialing/campaign management environments
    - Voice recognition as an enhanced IVR, ACD or unified messaging capability

  These functions are inherently supported in the same server architecture on the Interaction Center Platform, resulting in a greatly simplified architecture for deploying IP-based communication and related application services. Whereas IP-PBXs and softswitches have made it possible to run voice over IP (VoIP) and perform a number of call control and call management features, many IP-PBXs and softswitches are not

capable of handling media processing on the same platform and thus require adjunct servers.

- **A cost-effective migration path** from traditional circuit-based telephony to various IP-based telephony architectures such as SIP without redesigning applications. Applications built using the Interaction Center Platform run virtually unmodified regardless of whether the telephony service layer is circuit-based or IP-based. This allows organizations to migrate to IP telephony as they see fit while protecting current investments in applications.

- **Interoperability with other communication services** for full interaction management. Many of the software products built on the Interaction Center Platform provide out-of-the-box interfaces to back-end communication services such as e-mail, database, Web, and directory services that in many cases are already components of the installed IT infrastructure.

- **Single points of administration, customization, reporting and desktop control.** Whether circuit-based or IP-based, organizations utilizing Interaction Center Platform-based products are afforded common, inherent modules to manage system resources. For instance, they can use the *Interaction Administrator®* graphical console to configure lines, stations, users, workgroups, dial plans, etc. They can also use *Interaction Designer®* to build business rules for handling interactions. And agents, employees and other enterprise users alike can use the common *Interaction Client®* desktop interface to route all voice, fax, e-mail and Web interactions and generate reports across supported communications channels. These single points of administration, customization, reporting and desktop control combine to dramatically reduce administration and training costs throughout an entire organization.

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White Papers and SIP Data Sheet

For more information on Interactive Intelligence's IP telephony and SIP capabilities, read the **IP Telephony and the Interaction Center Platform** white paper. (1.5Mb PDF)

For more information on the Interaction Center Platform technology, read the **Interaction Center Platform** white paper. (552Kb PDF)

Downloadable **SIP data sheet** available in PDF format. (411Kb PDF)

The Facts About SIP: **Dispelling Fear, Uncertainty and Doubt** (164Kb PDF)

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JSRs: Java Specification Requests

JSR 116: SIP Servlet API

The SIP Servlet API defines a high-level extension API for SIP servers. It enables SIP applications to be deployed and managed based on the servlet model.

Status: Final

Stage		**Start**	**Finish**
Final Release	Download page	07 Mar, 2003	
Final Approval Ballot	View results	14 Jan, 2003	27 Jan, 2003
Proposed Final Draft	Download page	11 Dec, 2002	
Public Review 2	Download page	01 Oct, 2002	31 Oct, 2002
Public Review	Download page	10 Apr, 2002	09 May, 2002
Community Draft Reconsideration Ballot	View results	05 Feb, 2002	11 Feb, 2002
Community Draft Ballot	View results	18 Dec, 2001	24 Dec, 2001
Community Review	Login page	21 Nov, 2001	24 Dec, 2001
Expert Group Formation		17 Apr, 2001	
JSR Review Ballot	View results	03 Apr, 2001	16 Apr, 2001

JCP version in use: 2.1
Java Specification Participation Agreement version in use: 1.0
Please direct comments on this JSR to: jsr-116-comments@jcp.org

Specification Lead
Anders Kristensen   Dynamicsoft, Inc

Expert Group

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IBM	Nokia Networks	Siemens AG, ICN, CA IN E
Sun Microsystems, Inc.	Ubiquity	

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Original Java Specification Request (JSR)

Identification | Request | Contributions

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Section 2: Request

2.1 Please describe the proposed Specification:

The Session Initiation Protocol (SIP) is used to establish and manage multimedia IP sessions. This JSR requests the creation of a SIP Servlet API specification based on the model of the existing Servlet API. This will be version number 1.0.

2.2 What is the target Java platform? (i.e., desktop, server, personal, embedded, card, etc.)

Java™ 2 Standard Edition. We also expect the proposed specification to be a candidate for consideration to be part of the J2EE platform.

2.3 What need of the Java community will be addressed by the proposed specification?

The proposed API enables SIP servers to be augmented with Java extension

code which can be deployed and managed as a unit. The expert group will define an actual API as well as XML based deployment descriptors and related file formats. The SIP Servlet API is similar in nature to the HTTP Servlet API, but there are some important differences. HTTP servlets run only on origin servers. In SIP, proxy servers play a much bigger role, as do application servers which initiate requests. As such, SIP servlets must be able to run on these other server types.

The SIP Servlet API specification must address the following requirements:

- allow network servers to handle SIP requests in the delivery of SIP related services. This requires the ability to respond to requests, proxy requests, and initiate new requests.
- lifecycle management of SIP servlets and related SIP abstractions
- provide session management support, allowing users to deposit and retrieve data from objects which potentially span multiple SIP requests, calls, and even multiple protocols
- provide high level access to SIP objects, such as requests and responses, with the ability to manipulate key headers and field values. Emphasis is on simplicitly and minimality rather than completeness.
- must hide the complexities of SIP wherever possible; developers should not need to be SIP experts
- definition of rule based mappings from SIP requests to servlets which will process them
- definition of a security model
- definition of an XML DTD for SIP Servlet deployment descriptors
- definition of a jar based file format for SIP Servlet applications (similar to the Web ARchive (war) format defined for the HTTP Servlet API)
- possibly access to location databases

This list of requirements is not necessarily complete and neither will all items be addressed by version 1.0. In particular we do not expect to define SIP servlet mappings, deployment descriptors, and file formats in version 1.0. We intend to define the API in a progression of specifications which address the requirements in an incremental fashion. This is done to ensure timely delivery of the API as well as in order to gain experience with some of the more advanced features prior to standardization.

Please note that while the ability to proxy is central to SIP services, we make no assumption about whether this functionality is provided for servlets generally or for SIP servlets specifically.

2.4 Why isn't this need met by existing specifications?

The JAIN SIP API (JSR 32) defines a general purpose API which is intended for low-level SIP processing in clients as well as servers. Servlets add another layer of API to handle the specific needs of high volume servers that process SIP services developed by third parties. This introduces the need for many of the higher level services listed above: lifecycle management, sessions, data storage and retrieval, security, mappings, context and configuration data.

Where the JAIN SIP API exposes the full complexity of the SIP protocol, the servlet API will present an abstracted view of SIP to application writers and

will be designed to allow servlet engines to 1) handle those details of SIP protocol operation not needed for authoring services, and 2) prevent services from performing protocol violations or other restricted operations.

2.5 Please give a short description of the underlying technology or technologies:

SIP is an IETF protocol used for establishing, managing, and terminating sessions between two or more IP endpoints. It defines a number of network entities, notably user agents (UAs) which are the endpoints which initiate and respond to SIP requests, and proxies which makes routing decisions and forward SIP messages towards their destination UA. One of the main functions of SIP is routing session invitations from UA clients via a path of SIP proxies to UA servers. SIP servlets will typically reside on network servers where they will be responsible for making routing decisions.

The Servlet API was developed as an extension mechanism for Web servers similar to CGI scripts and native APIs. It allows Web servers to invoke Java based Web components in order to have them generate content dynamically. The Servlet API consists of a generic, protocol independent part, javax.servlet, as well as an HTTP specific part, javax.servlet.http. Apart from applying to SIP and not HTTP, the envisioned SIP Servlet API differs from the (HTTP) Servlet API primarily in its need to support proxying functionality and in having a more complicated state model.

2.6 Is there a proposed package name for the API Specification? (i.e., `javapi.something`, `org.something`, etc.)

javax.servlet.sip. The expert group will investigate whether changes to javax.servlet are desirable, and if so, will work with the servlet expert group to make this happen.

2.7 Does the proposed specification have any dependencies on specific operating systems, CPUs, or I/O devices that you know of?

No.

2.8 Are there any security issues that cannot be addressed by the current security model?

No.

2.9 Are there any internationalization or localization issues?

Similar to the HTTP Servlet API, e.g. servlet authors will be allowed to specify locale for responses.

2.10 Are there any existing specifications that might be rendered obsolete, deprecated, or in need of revision as a result of this work?

This is the first protocol other than HTTP for which an API is defined based on the javax.servlet package. It is possible that it turns out to be desirable to modify parts of the (HTTP) Servlet API, e.g. to support a common definition of sessions. The working group will investigate if modifications are needed, and if so, how they can be made in a way that ensures backwards compatibility of existing HTTP servlet code.

2.11 Please describe the anticipated schedule for the development of this specification.

Initiation: May 2001
Community Review: July 2001
Public Review: August 2001
Final Draft Proposal: October 2001

Section 3: Contributions

3.1 Please list any existing documents, specifications, or implementations that describe the technology. Please include links to the documents if they are publicly available.

M. Handley, H. Schulzrinne, E. Schooler, and J. Rosenberg, "SIP: Session Initiation Protocol", Nov. 2000
http://www.ietf.org/internet-drafts/draft-ietf-sip-rfc2543bis-02.txt

A. Deo, A. Kristensen, P. Mataga, K. Porter, J. Rosenberg, and P. Sripathi, "Overview of the SIP Servlet API", dynamicsoft Inc, Mar. 2001.